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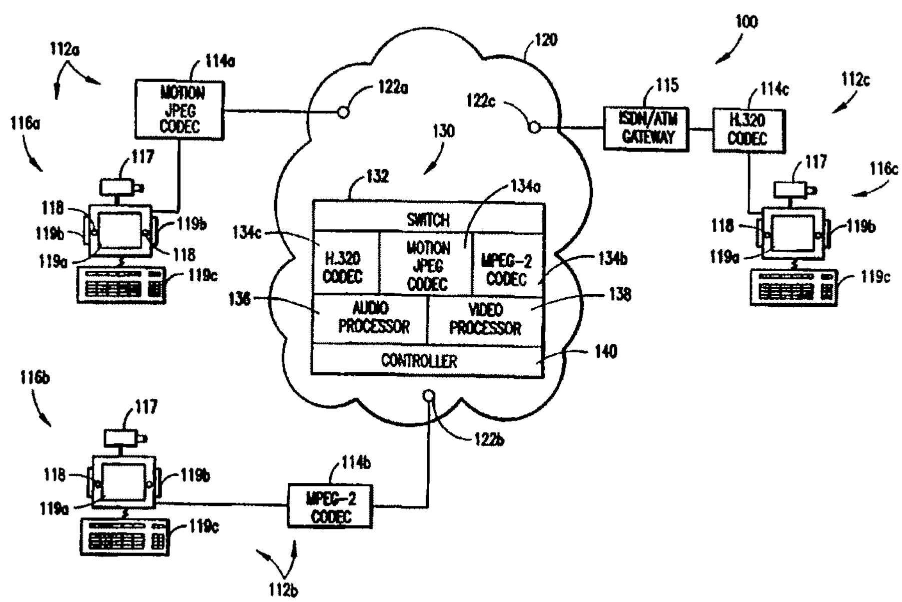

(57) Abstract

A multipoint multimedia server (130) includes a number of different compression codecs (134), a multipoint switch (132), separate audio (136) and video (138) processors, and at least one controller (140). Data streams of different compression standards enter the server (130) wherein the data streams are directed to the appropriate codec. The signals are mixed and switched by the controller (140) and the multipoint switch (132) and then routed back to the appropriate codecs. The signals are recompressed to the appropriate standard for each user before exiting the server (130).

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INTERCONNECTING MULTIMEDIA DATA STREAMS HAVING DIFFERENT COMPRESSED FORMATS

BACKGROUND OF THE INVENTION

1. Field of the Invention

The present invention relates broadly to multimedia telecommunications. More particularly, the present invention relates to methods, apparatus, and systems for the handling of compressed multimedia communication data so that multimedia equipment utilizing different data compression formats can be interconnected with each other.

2. State of the Art

With the advent of the optical network the telecommunications bandwidth available to business and individuals has increased dramatically. One concept utilizing the increased bandwidth is video communication (more commonly now called "multimedia" communications) which permits video, audio, and in some cases other data to be transported from one party to another or others. Multimedia communications can be utilized for a number of applications, and in different configurations. One configuration of recent interest has been multimedia conferencing, where several parties can communicate in a conference style.

In multimedia conferencing, the video data is handled such that each party can see at least one of the other parties, while the audio data is handled such that each party can hear one, several, or all of the other parties. In fact, various telecommunications standards are presently being adopted by the ITU-T and ISO which govern the protocols of multimedia conferencing (see, e.g., H.320, ITU-T.120). As part of the multimedia conferencing, various standards have been adopted for the compression of the video and audio data. For example, among compression standards applied to video compression are the JPEG (Joint Photographic Experts Group) standards promulgated by the joint ISO/CCITT technical committee ISO/IEC JTC1/SC2/WG10, and the MPEG (Motion Picture Experts Group) standards promulgated by ISO under ISO/IEC 11172(MPEG-1) and ISO/IEC 13818(MPEG-2). Another compression standard known as H.261 provides a very high compression ratio. Among the audio compression standards are the G722, G728 and MPEG audio compression. In addition, other compression techniques such as ADPCM (adaptive differential pulse code modulation) are known.

In the multimedia conferencing systems of the art (as represented by prior art Figure 1), the audio, video, and other data streams generated by a user's system 12a are multiplexed together directly in the encoder section of a multimedia encoder/decoder (codec) 14 located at the source/terminal 16, and transported together as an indivisible stream through the transport network 20 (now proposed in ATM format) to a similar "peer" codec 24 at a remote location. The peer codec is either at the remote user site for a point-to-point conference, and/or at a multimedia bridge 26 for a multipoint conference. The multimedia bridge 26, which typically includes a codec/switch 24 and a controller 28, provides conference control (e.g., it determines the signal to be sent to each participant), audio mixing (bridging) and multicasting, audio level detection for conference control, video switching, video mixing (e.g., a quad split, or "continuous presence device" which combines multiple images for display together) when available and/or desirable, and video multicasting. The audio and video data exiting the multimedia bridge is multiplexed, and continues through the transport network 20 to the desired multimedia source terminals 12b, 12c.

While the multimedia conference systems of the art are generally suitable for multimedia conferencing purposes, presently each terminal 12 connected to the conference must use a compatible codec 14. For example, a terminal using a Motion JPEG codec cannot communicate with a terminal using an MPEG or H.320 codec. Similarly, a terminal using an MPEG codec cannot communicate with a terminal using a Motion JPEG or H.320 codec, etc.

There have been recent proposals to incorporate "transcoders" within the fabric of an ATM network or at a user site. The proposed transcoders would have the ability to input a data stream, determine the compression standard being used, and, with the use of a transcoding algorithm, output a data stream which uses a different compression standard. To date, however, there has been little success in providing an acceptable transcoder. The algorithms necessary to convert one compression standard to another are difficult to design. The algorithms provided thus far have exhibited severe degradation of the output data stream. In addition, switching and mixing of multimedia data is rendered more difficult when using a transcoder.

SUMMARY OF THE INVENTION

It is therefore an object of the present invention to provide methods, apparatus, and systems for interconnecting compressed multimedia communication data streams having different compression formats.

It is another object of the invention to provide methods, apparatus, and systems for converting a data stream compressed with one standard to a data stream compressed with a different standard.

It is a further object of the invention to provide methods, apparatus, and systems which allow mixing and switching data from data streams compressed with different standards.

It is an additional object of the invention to provide methods, apparatus, and systems for converting a data stream compressed with one standard to a data stream compressed with a different standard wherein the output data stream signal quality is not substantially degraded.

In accord with these objects which will be discussed in detail below, the methods, apparatus, and systems according to the invention include providing a multipoint multimedia server which includes a plurality of different compression-scheme related codecs, a multipoint switch, separate audio and video processors, and at least one controller. Data streams of different compression standards from different users enter the server wherein the data streams are directed to the appropriate codec and are decoded to either baseband analog form, or more preferably to decompressed digital form. The decompressed digital or baseband analog signals are mixed and switched by the controller(s) and the multipoint switch according to user commands and are then routed back to appropriate codecs where the mixed and switched signals are recompressed to the appropriate standard for each user before exiting the server. The server is typically located at a remote node which is part of the communications network. The server, according to the invention preferably includes codecs for MPEG, Motion JPEG, and H.320 compression standards and may include other codecs based on proprietary schemes, as desired. While the presently preferred embodiment of the invention converts all of the data streams to analog audio and video prior to mixing and switching, in accord with an alternate embodiment, an appropriate digital multipoint switch can enable the data streams to be mixed and switched in decompressed (baseband) digital form. The multimedia terminals served by the invention may utilize NTSC, PAL, or SECAM video standards and the data streams may include other data as well as audio and video.

According to a preferred embodiment of the invention, the server is coupled to an ATM network. However, the server according to the invention may be used with other networks, either LAN or WAN.

Additional objects and advantages of the invention will become apparent to those skilled in the art upon reference to the detailed description taken in conjunction with the provided figures.

BRIEF DESCRIPTION OF THE DRAWINGS

Figure 1 is a high level diagram of a prior art multimedia conferencing system having a plurality of same-compression-type multimedia conferencing sites coupled by an ATM telecommunications network;

Figure 2 is a high level diagram of a multipoint multimedia server according to the invention coupled to a node of an ATM network serving a plurality of multimedia conferencing sites each utilizing a different data compression standard;

Figure 2a is a block diagram of the functionality of a portion of the server of Figure 2;

Figures 2b and 2c are flow diagrams of, respectively, the audio processor and video processor of the server of Figure 2a;

Figure 3a is a block diagram of a Motion JPEG codec used in the server according to the invention;

Figure 3b is a block diagram of an MPEG codec used in the server according to the invention;

Figure 3c is a block diagram of an H.320 codec used in the server according to the invention;

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENTS

Turning now to Figure 2, a first system 100 according to the invention services a plurality of users 112a, 112b, 112c, each of which are each provided with codecs 114a, 114b, 114c, and multimedia source/terminals 116a, 116b, 116c. The codecs 114c-114c, which are

described in greater detail with reference to Figures 3a-3c act as an interface between the network 120, and the source/terminals 116. The source/terminals typically include cameras 117, microphones 118, and computer terminals 119. The computer terminals 119, in turn, typically include a video monitor 119a, speakers 119b, keyboards 119c, etc. as is known in the art. As shown in Figure 2, user 112a is provided with a Motion JPEG codec 114a, user 112b is provided with an MPEG codec 114b, and user 112c is provided with an H.320 codec 114c. It will be appreciated that the network 120 is preferably an ATM network and, as such, the H.320 codec 114c is coupled to the network 120 via an ISDN/ATM gateway 115. It will further be appreciated that each of the users 112a-112c, is coupled to the network 120 at a node 122a, 122b, 122c. The system 100 also includes a multipoint multimedia server 130 which may be located anywhere in the ATM network 120.

The server 130 according to the invention, includes an ATM switch 132, a plurality of codecs 134a, 134b, 134c...134n, an audio processor 136, a video processor 138, and a controller 140. As shown, the codec 134a is a Motion JPEG codec, the codec 134b is an MPEG-2 codec, and the codec 134c is an H.320 codec. The codecs are described in detail below with reference to Figures 3a-3c. Generally, however, each codec receives an appropriate data stream via the switch 132 and passes a decoded signal (either baseband analog or decompressed digital) to the audio processor 136 and the video processor 138.

As seen in Fig. 2a, where the codecs provide baseband analog signals to the audio processor 136, the audio analog signals are converted by an A/D converter 136a into digital samples which are fed via a dedicated bus 136b to a DSP 136c. The digital samples are then processed by the DSP 136c (as discussed in more detail below with reference to Fig. 2b). For example, the audio information (samples) may be mixed under control of the controller 140 with the audio information of other users of the multimedia conference. Typically, the audio mixing involves summing, or weighting and summing the audio of all of the audio signals of the users in the conference, and then subtracting the source audio signal of the destination. After processing by the DSP 136c, the mixed audio samples are sent via the bus 136b for conversion to a baseband analog signal by a D/A converter 136d. The so-obtained mixed audio information may then be coded by the appropriate codec and forwarded to its destination via the ATM switch 132.

Where the codecs provide baseband analog video signals to the video processor 138, the video processor 138 may, under control of system controller 140, simply route (via switch 138a) the video information to the codecs, or may multiplex video information using the split processor 138b for split or multi screen transmission to its destination(s). Details of the video processing are discussed in more detail below with reference to Fig. 2c.

At each destination, the audio and video information being received is provided to the user's codec 114 in the proper compression format for processing, and then provided to the video monitor 119a and to the speaker 119b.

As seen in Fig. 2a, the server 130 also includes a data processing subsystem 137 which preferably utilizes an Ethernet interface between the codecs 134 and its T.12x stack 137a-137e. The data processing subsystem 137 can receive conference creation, scheduling, and management information which is passed to the system controller 140. As indicated, the system controller 140 provides various functions including node control and security 140a, operations, administration and management functions 140b, audio and video control 140c, and conference scheduling 140d. The audio and video control function 140c interfaces with the audio and video processors 136 and 138 as previously described.

Turning now to Fig. 2b, a flow chart of the audio processor function is seen. Thus, at 150, the audio processor initializes itself by setting up buffers, pointers, and assigning default register values. Then, at 152, the audio processor awaits commands from the system controller 140. Upon receiving at 154 a command, the audio processor DSP validates the command at 156 and processes the command at 158 and sets required flags for the interrupt service routine.

The interrupt service routine of the audio processor DSP 136c (entered at n/f intervals where n is the number of buffered audio samples per port and f is the sampling frequency) starts at 160 with the DSP obtaining from the A/D converters 136a the audio samples available at all ports and storing them into a matrix S. At 162, for each port P which is inactive or muted, all samples for that port are set to zero in the matrix; i.e., $S[P] = 0$. For all of the ports relating to a conference (i.e., a group), the samples are added together at 164; i.e., $T(g) = \sum S[P]$ for the ports of that group. At 166, the power $P[p]$ of each port is computed. Then, at 168, for each conference, the loudest port $L[p]$ is found by comparing the powers $P[p]$ for that group. At 170, for each port which is subject to an automatic gain control, the gain is adjusted according to the effective and targeted power values as described in co-owned, copending U.S. Patent Application SN 08/888,571 to Tabet et al., filed July 7, 1997, which is hereby incorporated by reference herein in its entirety. At 172, for each port, the sample vector for that port is subtracted from the sum of the group vectors; i.e., $S[P] = T[g] - S[P]$, to provide a mixed audio sample for output. The mixed audio samples are then sent out at 174 for digital to analog conversion.

Turning to Fig. 2c, a flow chart of the video processing function is provided. At 180, the video system initializes itself. At 182, the video system awaits a command from the controller 140. Upon receiving at 184 a command, the video switch is formatted at 186 such

that any input (whether from the codecs or the split processor 138b) can be switched to any output (including the codecs and the split processor).

A high level block diagram of the Motion JPEG codec 114a (134a) is seen in Figure 3a. The Motion JPEG codec generally includes audio circuitry 200, video circuitry 220, modem type data circuitry 240, and a main processor or host 250 with an associated bus 252 and associated circuitry such as a boot PROM 254, flash memory 256, SRAM 258, a local manager interface 260, a timer 262, a watchdog circuit 264, and a management port UART 266. The boot PROM 254 stores the code which boots the main processor 250. The flash memory 256 is typically utilized to hold the main code and static configuration information. The RAM 258 is typically a dynamic RAM for running the code and temporarily storing data. The timer 262 provides clock signals for the operating system, while the watchdog 264 performs reset functions. The management port UART 266 is provided for access by a system manager (not shown) to the codec, while the local management interface 260 provides the codec with the capability of interfacing with a local manager such as the controller 140 of Figure 2.

The audio circuitry 200 includes an analog to digital converter 202, an audio codec interface 204, an audio source time manager 206, an audio packet buffer SRAM 210, a packet buffer and DMA controller 212, an audio channel output interface 214, an audio channel input interface 216, an audio presentation time manager 217, and a digital to analog converter 218. The video circuitry 220 includes a composite NTSC/PAL video decoder 222, a JPEG compressor 224, a video source time manager 226, an outgoing packet buffer SRAM 227, an outgoing video packet buffer and DMA controller 228, a video channel output interface 229, and a video channel input interface 231, an incoming packet buffer and DMA controller 232, an incoming packet buffer SRAM 233, a JPEG decompressor 234, a composite NTSC/PAL video encoder 236, a video presentation time manager 238, and a video sync generator 239. The data circuitry 240, includes a data channel port UART 242, a data packet buffer and DMA controller 244 with an associated data packet buffer RAM 245, a data channel output interface 246, and a data channel input interface 248.

Generally, outgoing audio information received from a microphone(s) or other audio source is applied to the analog to digital converter 202 which simultaneously provides the digital audio data to the audio codec interface 204 and, in accord with a preferred aspect of the invention, provides a reference clock to the audio source time manager 206. The audio codec interface 204 converts the format of the data received from the A/D converter so that the data may be properly provided to the packet buffer SRAM 210 under control of the packet buffer and DMA (direct memory access) controller. In addition, in accord with the preferred embodiment, the main processor 250 provides a PES (Program Elementary Stream) header to the SRAM 210

to effectively generate PES formatted packet. The packet buffer and DMA controller 212 controls the movement of the packetized audio data from the SRAM 210 to the channel output interface 214 as required. The channel output interface 214, in turn, places the data in a desired format (e.g., a Transport Stream (TS) format, or an ATM format) by inserting a system time indicator (provided by the audio source time manager) into the signal, and provides the desired overhead bits or bytes (including OAM where appropriate). The channel output interface 214 implements a serial channel physical interface by receiving the parallel stream of data from the buffer controller, and converting the parallel stream into a serial stream with an accompanying clock, etc., which is multiplexed with video data by the multiplexer/demultiplexer 270.

Incoming audio information is received by the audio channel input interface 216 from the multiplexer/demultiplexer 270. The audio channel input interface 216 frames on the incoming (TS) cell, checks the headers for errors, passes the payload in byte wide format (parallel format) to the packet buffer and DMA controller 212, and passes a time reference marker (Program Clock Reference value -PCR) to the audio presentation time manager 217. The DMA controller 212 places the payload in desired locations in the SRAM 210. When the data is to be presented to the receiver as audio information, the DMA controller takes the data out of the SRAM 210 and provides it to the audio codec interface 204, which reformats the data into a serial stream for digital to analog conversion by the D/A converter 218. The presentation time manager 217 is provided to recover a local clock.

The video circuitry 220 processes and outputs the video signals. Outgoing video information is received by the video circuitry 220 as a composite analog input. The composite input is decoded by the composite decoder 222 which provides digital luminance and color difference signals to the video compressor 224, and horizontal and vertical synch and a field indicator to the video source time manager 226. The Motion JPEG video compressor 224 compresses the data, and generates a JPEG frame with start of image and end of image markers. The video compressor 224 puts the framed compressed data in parallel format, so that the buffer controller 228 can place the compressed data into the packet buffer SRAM 227. Preferably, the host (main processor 250) provides PES headers via the buffer controller 228 to desired locations in the SRAM 227 to effectively convert the JPEG frame into a PES packet. The packet buffer and DMA controller 228 provides the channel output interface 229 with the PES packet data at a constant rate. If sufficient data is not available in the packet buffer SRAM 227, the channel output interface 229 generates an "idle cell". Regardless, the channel output interface 229, places the data in a desired format (e.g., TS or ATM format) by inserting a system time indicator (provided by the video source time manager 226) into the signal, and provides the desired overhead bits or bytes (including OAM where appropriate). The channel output interface 229 implements a serial channel physical interface by receiving the parallel

stream of data from the buffer controller 228, and converting the parallel stream into a serial stream with an accompanying clock, etc. The outgoing stream of video data is multiplexed with time-related audio data at the multiplexer/demultiplexer 270.

In the receiving direction, video data which is demultiplexed by the multiplexer/demultiplexer 270 is obtained at the video channel input interface 231 which frames on the incoming (TS) cell, checks the headers for errors, passes the payload in byte wide format (parallel format) to the packet buffer and DMA controller 232, and passes a time reference marker to the video presentation time manager 238. The DMA controller 232 places the payload in desired locations in the SRAM 233. When the JPEG decompressor 234 indicates that the next video field is required for display, the DMA controller 232 provides the buffered compressed video from the head of the buffer to the JPEG decompressor 234. The JPEG decompressor 234, in turn, decompresses the data, and provides digital luminance and color difference signals to the composite video encoder 236.

The composite video encoder 236 operates based on the video timing signals (horizontal line timing or H, vertical field timing or V, and the field indicator) generated by the sync generator, based on the sample clock recovered by the presentation time manager from the channel PCR. The composite video encoder 236 in turn indicates to the JPEG decompressor when it requires video data for the next video field, and which field is required. Based on these timing signals and the decompressed video data, the composite video encoder generates the analog video output for the video monitor 119a or for the video processor 138.

Turning now to Figure 3b, an MPEG-2 codec 114b, 134b generally includes audio circuitry 300, video circuitry 320, controller circuitry 340, management port circuitry 350, multiplexer/demultiplexer circuitry 360, and channel interface circuitry 370. The audio circuitry 300 includes an analog to digital converter 302, a digital to analog converter 304, an MPEG-2 audio processor (compressor/decompressor) 306, and an audio buffer controller 308. The video circuitry 320 includes a composite NTSC/PAL video decoder 322, an MPEG-2 compressor 324, an outgoing video buffer controller 326, an incoming video buffer controller 328, an MPEG decompressor 330, and a composite NTSC/PAL video encoder 332. The controller circuitry 340 is coupled to the buffer controllers 308, 326, 328 as well as to the management port circuitry 350. The multiplexer/demultiplexer circuitry 360 is coupled to the buffer controllers 308, 326, 328 as well as to the channel interface circuitry 370.

Generally, incoming MPEG-2 coded audio/video data is received via the channel interface 370 and split into separate audio and video signal streams by the multiplexer/demultiplexer circuitry 360. The incoming audio data is passed to the audio buffer

controller 308 which provides the audio data to the MPEG-2 audio processor 306. The MPEG-2 audio processor 306 decompresses the audio data and provides baseband (decompressed) digital data which is converted by the digital to analog converter 304 into an analog audio output. The incoming video data is likewise passed to the video decoder buffer controller 328 which provides the video data to the MPEG-2 video decoder 330. The MPEG-2 video decoder 330 decompresses the video data and provides decompressed digital data which is converted by the video encoder 332 into a composite video output.

On the outgoing side, outgoing audio information received from a microphone(s) or other audio source (e.g. an audio mixer at the server) is applied to the analog to digital converter 302 which provides a digital output to the MPEG-2 audio processor 306. The MPEG-2 audio processor 306 in turn provides compressed digital audio data to the audio buffer controller 308. Likewise, the outgoing video data received from a camera or other video source (e.g. a video switch or mixer) in the form of composite video is applied to the Video decoder 312 which provides a digital signal to the MPEG-2 video encoder 324. The compressed digital data output of the MPEG-2 encoder 324 is passed through the buffer controller 326 to the multiplexer/demultiplexer 360 where the compressed video data is mixed with the compressed audio data for transmission via the channel interface 370 to the ATM network.

Turning now to Figure 3c, an H.320 codec 114c, 134c generally includes a video interface 402, 404, an audio interface 406, 408, a data port 410, a slot controller interface 412, a video compressor 414, a video decompressor 416, an audio compressor 418, an audio decompressor 420, a multiplexer 422 / demultiplexer 424, and a host processor 426. The video interface includes an NTSC or PAL video to CCIR-601 format decoder 402 and a CCIR-601 to NTSC or PAL encoder 404. The audio interface includes an A/D converter 406 and a D/A converter 408. The sampling rates of the converters 406, 408 are preferably programmable from 8KHz to 48KHz with either 8 or 16 bit samples. The data port is preferably an RS-232 interface 410. The slot controller interface 412 includes the same CSP as the Motion JPEG codec shown and described with reference to Figure 3a. The interface 412 also includes circuitry for formatting/deformatting an H.320 data stream flowing through an ATM adaptation processor. The video compressor 414 and decompressor 416 support the CCITT H.261 (full CIF and QCIF) recommendation for video codecs. This provides a video frame rate of 15 fps with an H.320 serial speed of 384 Kbps. The audio compressor 418 and decompressor 420 support G.711 (μ-law, A-law, PCM), G.722 (ADPCM), and G.728 (CELP) speech codecs and are preferably implemented with a DSP. The multiplexer 422 and demultiplexer 424 support the H.221 portion of the H.320 standard. Video, audio, data, and signalling are provided in a signals stream of 64 Kbps with the upper bound being 384 Kbps. The host processor 426 controls the H.320 codec and is preferably a Power PC.

As mentioned above with reference to Figure 2, each of the codecs in the server 130 can be arranged to provide either baseband analog or decompressed digital signals to the audio processor and the video processor for mixing and multiplexing according to user commands. From the foregoing description of the codecs, it will be appreciated that if the audio and video processing is to be performed on decompressed digital signals, the various A/D, D/A, and video coder/decoders described in the codecs may be omitted or bypassed.

There have been described and illustrated herein methods, apparatus, and systems for interconnecting compressed multimedia communication data streams having different compression formats. While particular embodiments of the invention have been described, it is not intended that the invention be limited thereto, as it is intended that the invention be as broad in scope as the art will allow and that the specification be read likewise. Thus, while the invention has been described with reference to three particular compression standards (Motion JPEG, MPEG-2, and H.320), it will be appreciated that it may be appropriate in some circumstances to include other standards with appropriate codecs provided that appropriate decompressed digital or baseband analog signals can be generated, or to exclude one of the disclosed codecs. It will also be appreciated that while the multimedia server of the invention has been described with reference to an ATM network and an ISDN network, other network architectures could be utilized. It will therefore be appreciated by those skilled in the art that yet other modifications could be made to the provided invention without deviating from its spirit and scope as so claimed.

Claims:

1. A telecommunications multimedia communications system for use in conjunction with a telecommunications network, said system comprising:

a) a first multimedia terminal coupled to the telecommunications network, said first multimedia terminal including a first terminal codec means for generating an outgoing audio/video data stream and for receiving an incoming audio/video data stream, said first terminal codec means utilizing a first compression standard;

b) a second multimedia terminal coupled to the telecommunications network, said second multimedia terminal including a second terminal codec means for generating an outgoing audio/video data stream and for receiving an incoming audio/video data stream, said second terminal codec means utilizing a second compression standard;

c) a multimedia server means coupled to the network for interconnecting said first and second terminal codec means, said server means including

i) a first server codec means for generating an outgoing audio/video data stream and for receiving an incoming audio/video data stream, said first server codec means utilizing said first compression standard,

ii) a second server codec means for generating an outgoing audio/video data stream and for receiving an incoming audio/video data stream, said second server codec means utilizing said second compression standard,

iii) audio processing means coupled to said first server codec means and to said second server codec means for processing audio signals received from said first server codec means and said second server codec means and for providing processed audio signals to said first server codec means and to said second server codec means,

iv) video processing means coupled to said first server codec means and to said second server codec means for processing video signals received from said first server codec means and said second server codec means and for providing processed video signals to said first server codec means and to said second server codec means, and

v) controller means coupled to said first server codec means, said second server codec means, said audio processing means, and said video processing means such that

video signals received from said first server codec means are, after processing by said video processing means, directed to said second server codec means for transmission to said second terminal codec means,

video signals received from said second server codec means are, after processing by said video processing means, directed to said first server codec means for transmission to said first terminal codec means,

audio signals received from said first server codec means are, after processing by said audio processing means, directed to said second server codec means for transmission to said second terminal codec means, and

audio signals received from said second server codec means are, after processing by said audio processing means, directed to said first server codec means for transmission to said first terminal codec means.

2. A telecommunications multimedia communications system according to claim 1, wherein:

said first and second server codec means provide decompressed digital audio data to said audio processing means.

3. A telecommunications multimedia communications system according to claim 1, wherein:

said first and second server codec means provide decompressed digital video data to said video processing means.

4. A telecommunications multimedia communications system according to claim 2, wherein:

said first and second server codec means provide decompressed digital video data to said video processing means.

5. A telecommunications multimedia communications system according to claim 1, wherein:

said first and second server codec means provide baseband analog audio data to said audio processing means.

6. A telecommunications multimedia communications system according to claim 1, wherein:

said first and second server codec means provide baseband analog video data to said video processing means.

7. A telecommunications multimedia communications system according to claim 5, wherein:

said first and second server codec means provide baseband analog video data to said video processing means.

8. A telecommunications multimedia communications system according to claim 1 where the telecommunications network is an ATM network, wherein:

said multimedia server means includes an ATM switch.

9. A telecommunications multimedia communications system according to claim 1, wherein:
said first compression standard is one of Motion JPEG, MPEG-2, and H.320, and
said second compression standard is another of Motion JPEG, MPEG-2, and H.320.

10. A telecommunications multimedia communications system according to claim 1, further comprising:
d) a third multimedia terminal coupled to the telecommunications network, said third multimedia terminal including a third terminal codec means for generating an outgoing audio/video data stream and for receiving an incoming audio/video data stream, said third terminal codec means utilizing a third compression standard; wherein
said multimedia server means further includes
vi) a third server codec means for generating an outgoing audio/video data stream and for receiving an incoming audio/video data stream, said third server codec means utilizing said third compression standard,
said audio processing means coupled to said third server codec means for processing audio signals received from said third server codec means and for providing processed audio signals to said third server codec means,
said video processing means coupled to said third server codec means for processing video signals received from said third server codec means and for providing processed video signals to said third server codec means, and
said controller means coupled to said third server codec means, such that
video signals received from said first and second server codec means are, after processing by said video processing means, directed to said third server codec means for transmission to said third terminal codec means,
video signals received from said third server codec means are, after processing by said video processing means, directed to said first and second server codec means for transmission to said first and second terminal codec means,
audio signals received from said first and second server codec means are, after processing by said audio processing means, directed to said third server codec means for transmission to said third terminal codec means, and
audio signals received from said third server codec means are, after processing by said audio processing means, directed to said first and second server codec means for transmission to said first and second terminal codec means.

11. A telecommunications multimedia communications system according to claim 10, wherein:
said first compression standard is Motion JPEG,
said second compression standard is MPEG-2, and
said third compression standard is H.320.

12. A multipoint multimedia server for use in conjunction with a telecommunications network having a first multimedia terminal coupled to the telecommunications network via a first terminal codec utilizing a first compression standard and a second multimedia terminal coupled to the telecommunications network via a second terminal codec utilizing a second compression standard, said server comprising:

a) a first server codec means for generating an outgoing audio/video data stream and for receiving an incoming audio/video data stream, said first server codec means utilizing the first compression standard,

b) a second server codec means for generating an outgoing audio/video data stream and for receiving an incoming audio/video data stream, said second server codec means utilizing the second compression standard,

c) audio processing means coupled to said first server codec means and to said second server codec means for processing audio signals received from said first server codec means and said second server codec means and for providing processed audio signals to said first server codec means and to said second server codec means,

d) video processing means coupled to said first server codec means and to said second server codec means for processing video signals received from said first server codec means and said second server codec means and for providing processed video signals to said first server codec means and to said second server codec means, and

e) controller means coupled to said first server codec means, said second server codec means, said audio processing means, and said video processing means such that

video signals received from said first server codec means are, after processing by said video processing means, directed to said second server codec means for transmission to the second terminal codec means,

video signals received from said second server codec means are, after processing by said video processing means, directed to said first server codec means for transmission to the first terminal codec means,

audio signals received from said first server codec means are, after processing by said audio processing means, directed to said second server codec means for transmission to the second terminal codec means, and

audio signals received from said second server codec means are, after processing by said audio processing means, directed to said first server codec means for transmission to the first terminal codec means.

13. A multipoint multimedia server according to claim 12, wherein:

said first and second server codec means provide decompressed digital audio data to said audio processing means.

14. A multipoint multimedia server according to claim 12, wherein:

said first and second server codec means provide decompressed digital video data to said video processing means.

15. A multipoint multimedia server according to claim 13, wherein:

said first and second server codec means provide decompressed digital video data to said video processing means.

16. A multipoint multimedia server according to claim 12, wherein:

said first and second server codec means provide baseband analog audio data to said audio processing means.

17. A multipoint multimedia server according to claim 12, wherein:

said first and second server codec means provide baseband analog video data to said video processing means.

18. A multipoint multimedia server according to claim 12 where the telecommunications network is an ATM network, said server further comprising:

f) an ATM switch coupled to said first and second server codec means for coupling said server to the network.

19. A multipoint multimedia server according to claim 12, wherein:

said first compression standard is one of Motion JPEG, MPEG-2, and H.320, and

said second compression standard is another of Motion JPEG, MPEG-2, and H.320.

20. A multipoint multimedia server according to claim 8 where the network includes a third multimedia terminal coupled to the network via a third terminal codec utilizing a third compression standard, said server further comprising:

f) a third server codec means for generating an outgoing audio/video data stream and for receiving an incoming audio/video data stream, said third server codec means utilizing the third compression standard,

said audio processing means coupled to said third server codec means for processing audio signals received from said third server codec means and for providing processed audio signals to said third server codec means,

said video processing means coupled to said third server codec means for processing video signals received from said third server codec means and for providing processed video signals to said third server codec means, and

said controller means coupled to said third server codec means, such that

video signals received from said first and second server codec means are, after processing by said video processing means, directed to said third server codec means for transmission to the third terminal codec means,

video signals received from said third server codec means are, after processing by said video processing means, directed to said first and second server codec means for transmission to the first and second terminal codec means,

audio signals received from said first and second server codec means are, after processing by said audio processing means, directed to said third server codec means for transmission to the third terminal codec means, and

audio signals received from said third server codec means are, after processing by said audio processing means, directed to said first and second server codec means for transmission to the first and second terminal codec means.

21. A multipoint multimedia server according to claim 20, wherein:

said first compression standard is Motion JPEG,

said second compression standard is MPEG-2, and

said third compression standard is H.320.

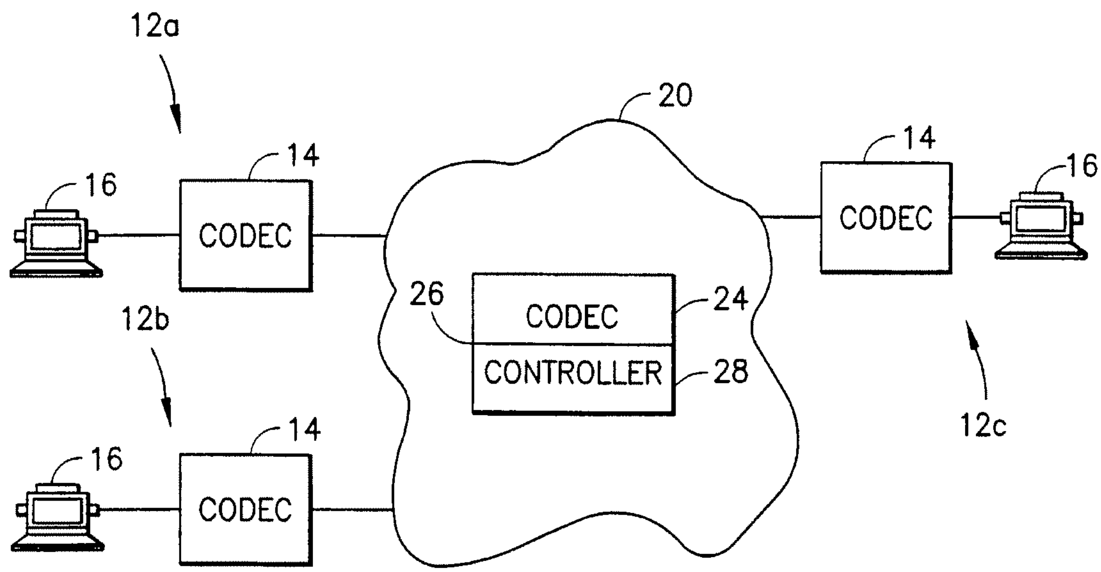


FIG. 1
PRIOR ART

2/10

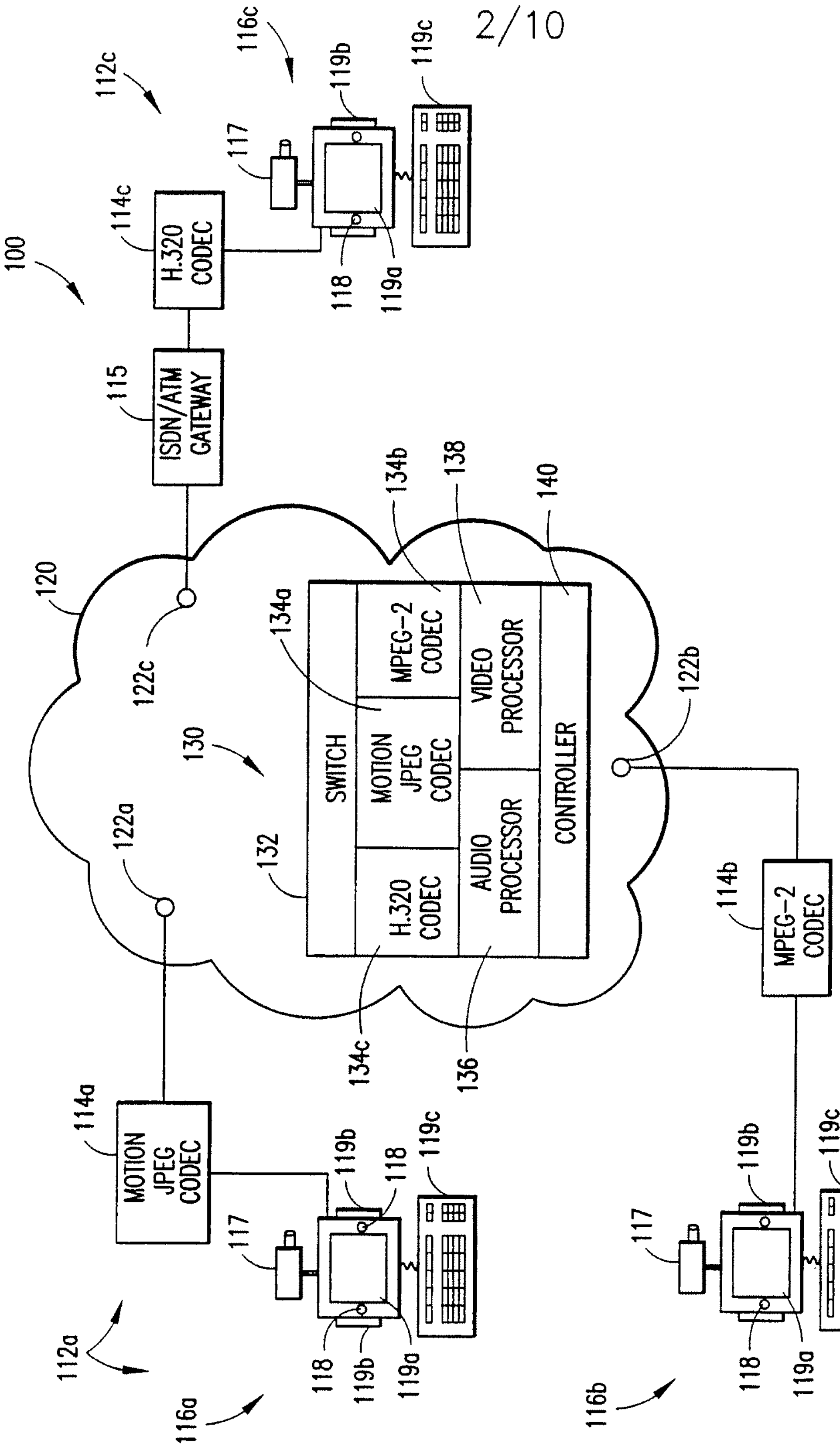

100
112a
112b
112c
114a
MOTION JPEG CODEC
114b
MPEG-2 CODEC
114c
H.320 CODEC
115
ISDN/ATM GATEWAY
116a
116b
116c
117
118
119a
119b
119c
120
122a
122b
122c
130
132
SWITCH
134a
MOTION JPEG CODEC
134b
MPEG-2 CODEC
134c
H.320 CODEC
136
AUDIO PROCESSOR
138
VIDEO PROCESSOR
140
CONTROLLER

FIG.2

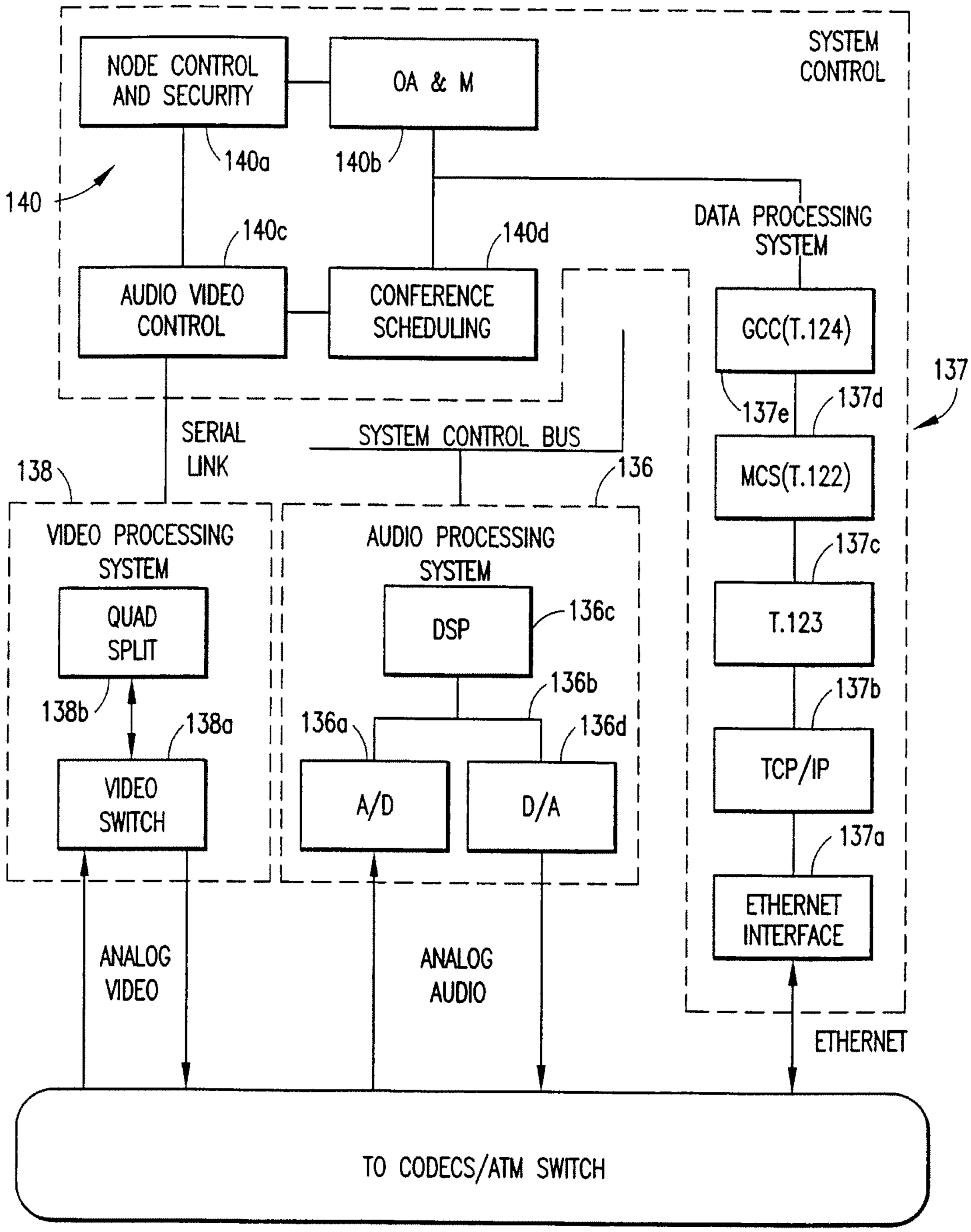
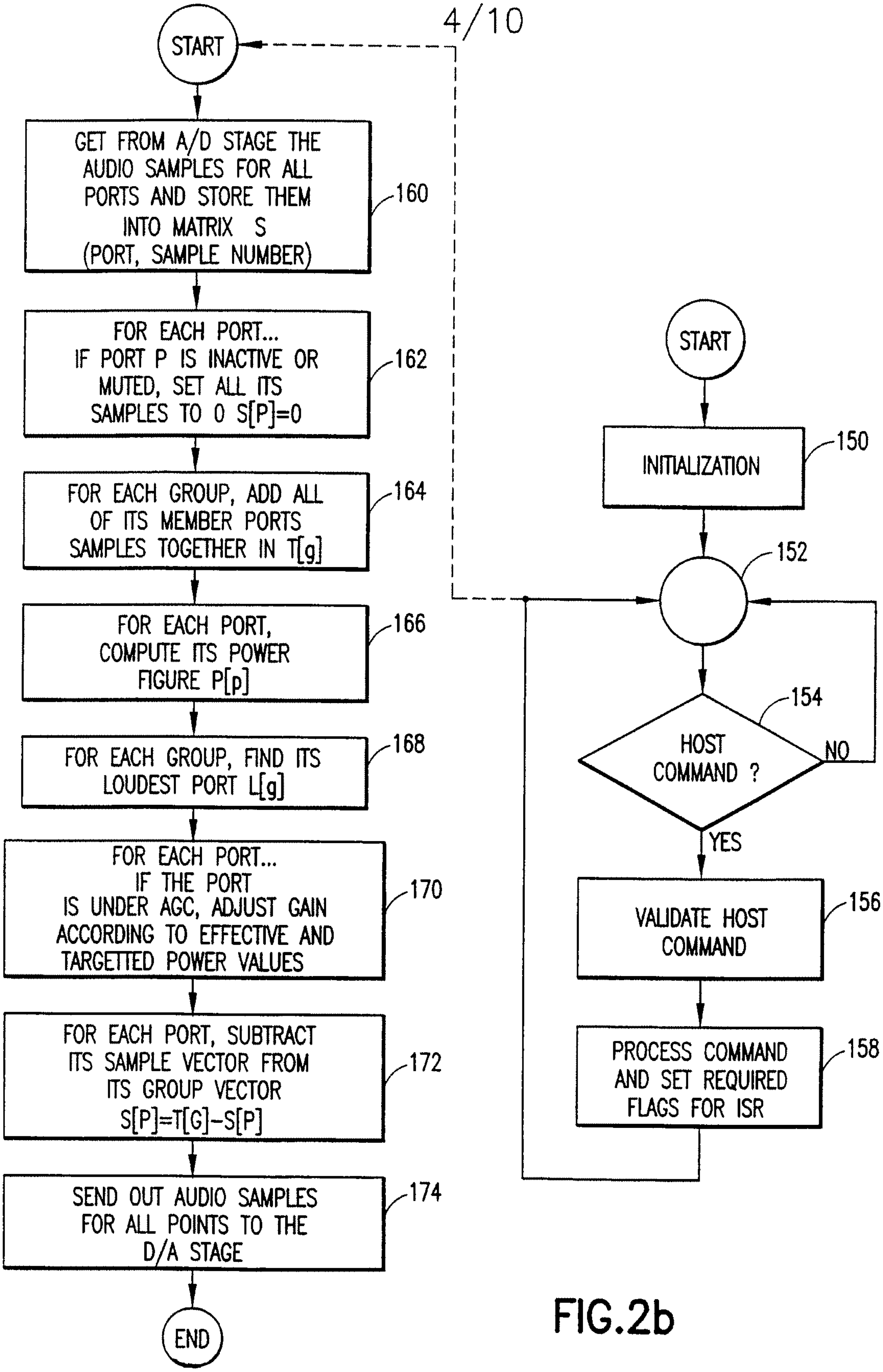

SYSTEM CONTROL
NODE CONTROL AND SECURITY
OA & M
140a
140b
140
140c
140d
DATA PROCESSING SYSTEM
AUDIO VIDEO CONTROL
CONFERENCE SCHEDULING
GCC(T.124)
137
137e
137d
SERIAL LINK
SYSTEM CONTROL BUS
138
136
MCS(T.122)
VIDEO PROCESSING SYSTEM
AUDIO PROCESSING SYSTEM
137c
QUAD SPLIT
DSP
136c
T.123
138b
136b
137b
138a
136a
136d
VIDEO SWITCH
A/D
D/A
TCP/IP
137a
ETHERNET INTERFACE
ANALOG VIDEO
ANALOG AUDIO
ETHERNET
TO CODECS/ATM SWITCH

FIG.2a



START
GET FROM A/D STAGE THE AUDIO SAMPLES FOR ALL PORTS AND STORE THEM INTO MATRIX S (PORT, SAMPLE NUMBER)
160
FOR EACH PORT... IF PORT P IS INACTIVE OR MUTED, SET ALL ITS SAMPLES TO 0 S[P]=0
162
FOR EACH GROUP, ADD ALL OF ITS MEMBER PORTS SAMPLES TOGETHER IN T[g]
164
FOR EACH PORT, COMPUTE ITS POWER FIGURE P[p]
166
FOR EACH GROUP, FIND ITS LOUDEST PORT L[g]
168
FOR EACH PORT... IF THE PORT IS UNDER AGC, ADJUST GAIN ACCORDING TO EFFECTIVE AND TARGETTED POWER VALUES
170
FOR EACH PORT, SUBTRACT ITS SAMPLE VECTOR FROM ITS GROUP VECTOR S[P]=T[G]-S[P]
172
SEND OUT AUDIO SAMPLES FOR ALL POINTS TO THE D/A STAGE
174
END
START
INITIALIZATION
150
152
HOST COMMAND ?
154
NO
YES
VALIDATE HOST COMMAND
156
PROCESS COMMAND AND SET REQUIRED FLAGS FOR ISR
158

FIG.2b

5/10

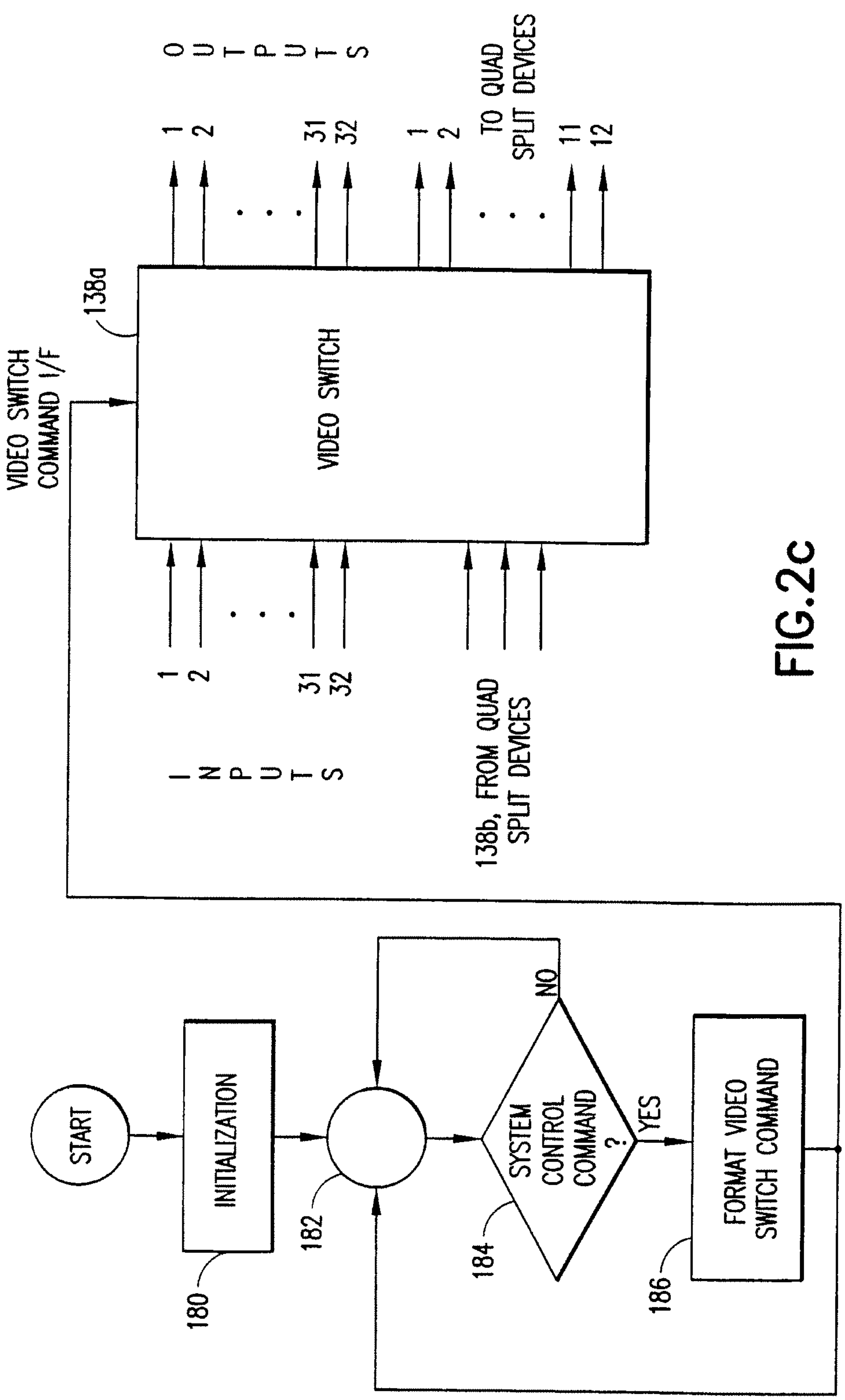

VIDEO SWITCH
COMMAND I/F
138a
VIDEO SWITCH
OUTPUTS
1
2
31
32
1
2
TO QUAD
SPLIT DEVICES
11
12
INPUTS
1
2
31
32
138b, FROM QUAD
SPLIT DEVICES
START
INITIALIZATION
180
182
SYSTEM
CONTROL
COMMAND
?
184
NO
YES
FORMAT VIDEO
SWITCH COMMAND
186

FIG.2c

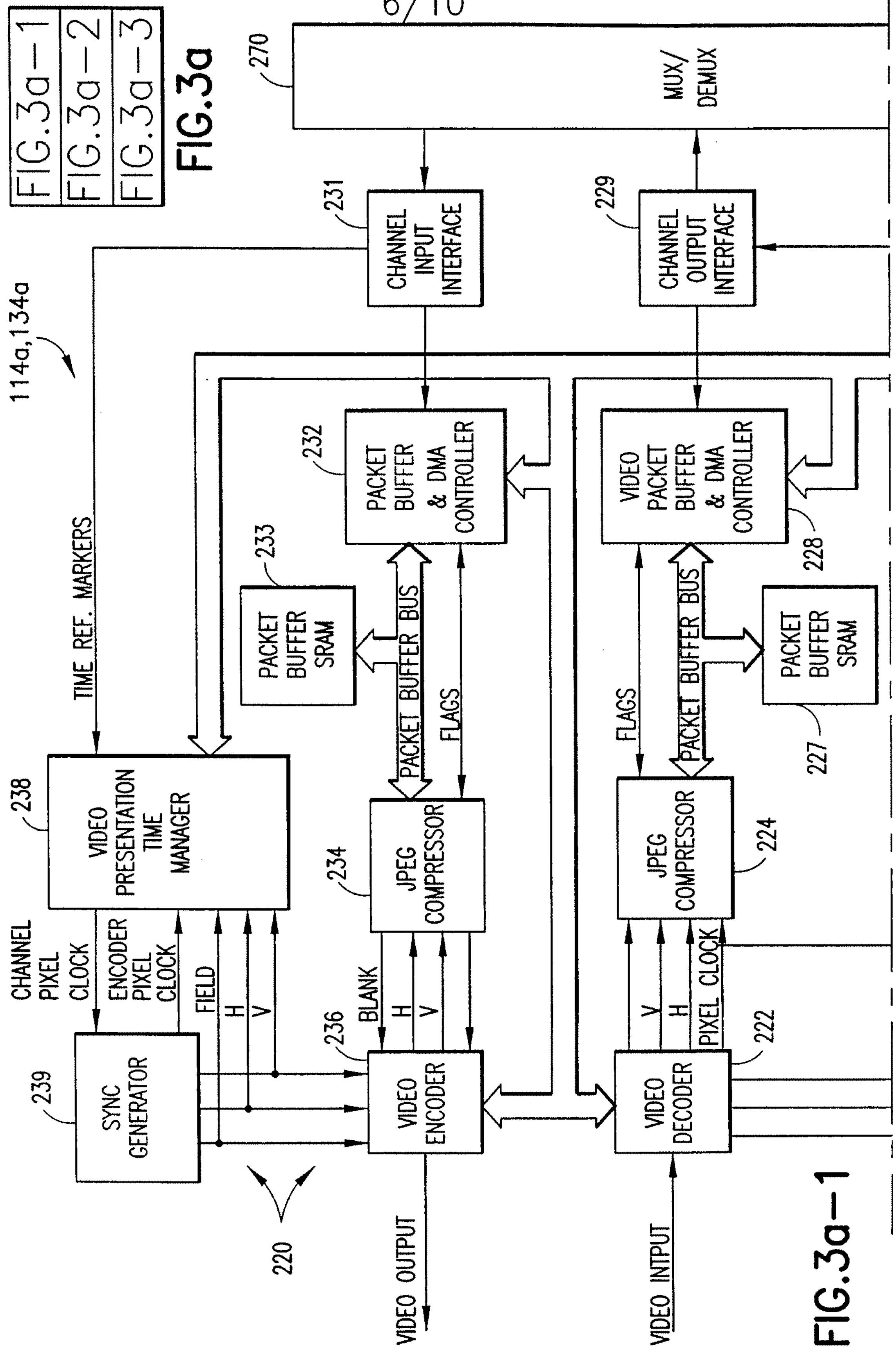


FIG.3a-1
FIG.3a-2
FIG.3a-3
FIG.3a
114a,134a
270
MUX/
DEMUX
231
CHANNEL
INPUT
INTERFACE
229
CHANNEL
OUTPUT
INTERFACE
TIME REF. MARKERS
232
PACKET
BUFFER
& DMA
CONTROLLER
VIDEO
PACKET
BUFFER
& DMA
CONTROLLER
228
233
PACKET
BUFFER
SRAM
PACKET BUFFER BUS
FLAGS
PACKET BUFFER BUS
FLAGS
PACKET
BUFFER
SRAM
227
238
VIDEO
PRESENTATION
TIME
MANAGER
234
JPEG
COMPRESSOR
JPEG
COMPRESSOR
224
CHANNEL
PIXEL
CLOCK
ENCODER
PIXEL
CLOCK
FIELD
H
V
BLANK
H
V
V
H
PIXEL CLOCK
239
SYNC
GENERATOR
236
VIDEO
ENCODER
222
VIDEO
DECODER
220
VIDEO OUTPUT
VIDEO INTPUT

FIG.3a-1

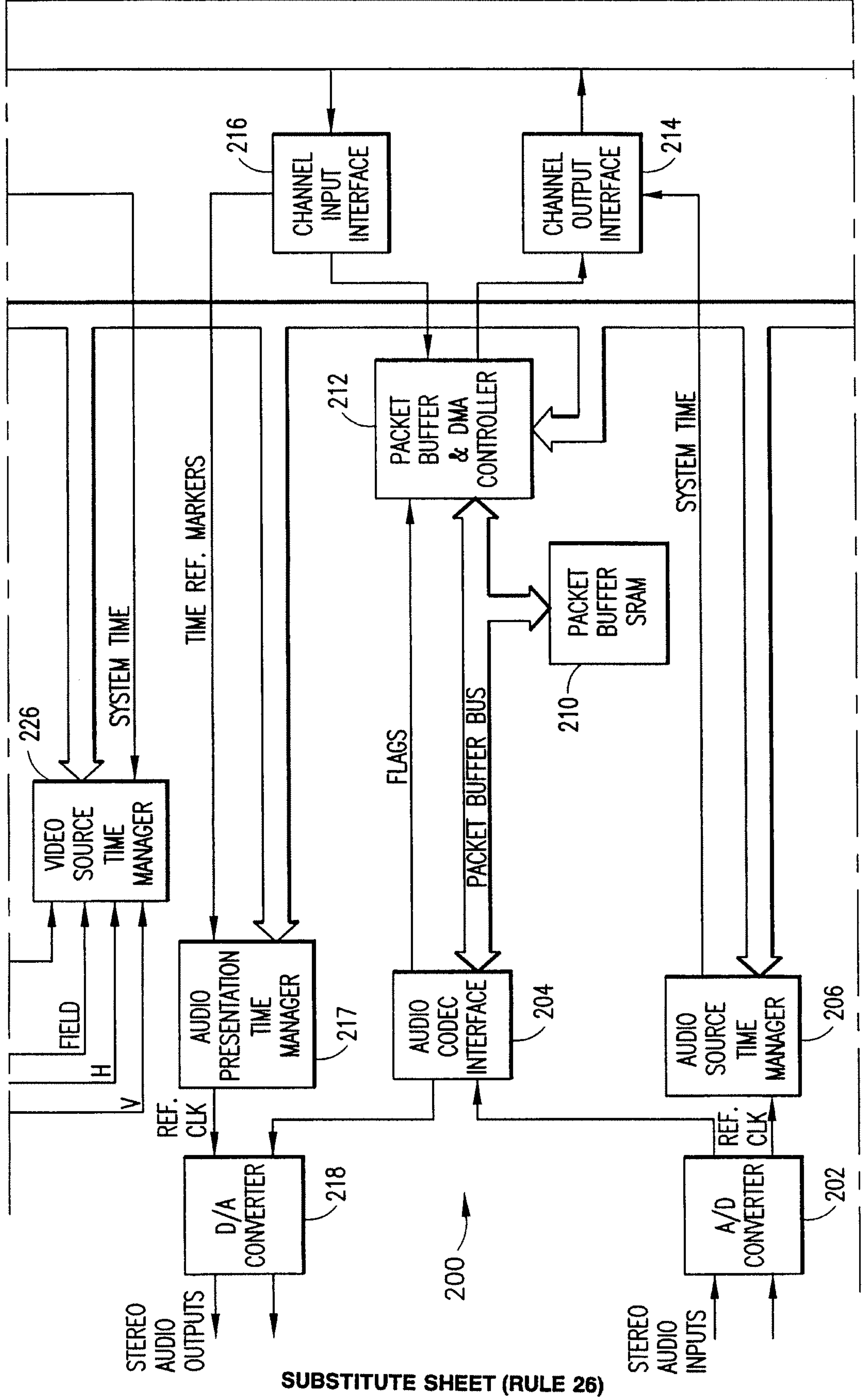

CHANNEL INPUT INTERFACE
216
CHANNEL OUTPUT INTERFACE
214
PACKET BUFFER & DMA CONTROLLER
212
PACKET BUFFER SRAM
210
SYSTEM TIME
TIME REF. MARKERS
SYSTEM TIME
VIDEO SOURCE TIME MANAGER
226
FLAGS
PACKET BUFFER BUS
FIELD
H
V
AUDIO PRESENTATION TIME MANAGER
217
AUDIO CODEC INTERFACE
204
AUDIO SOURCE TIME MANAGER
206
REF. CLK
REF. CLK
D/A CONVERTER
218
A/D CONVERTER
202
200
STEREO AUDIO OUTPUTS
STEREO AUDIO INPUTS

FIG.3a–2

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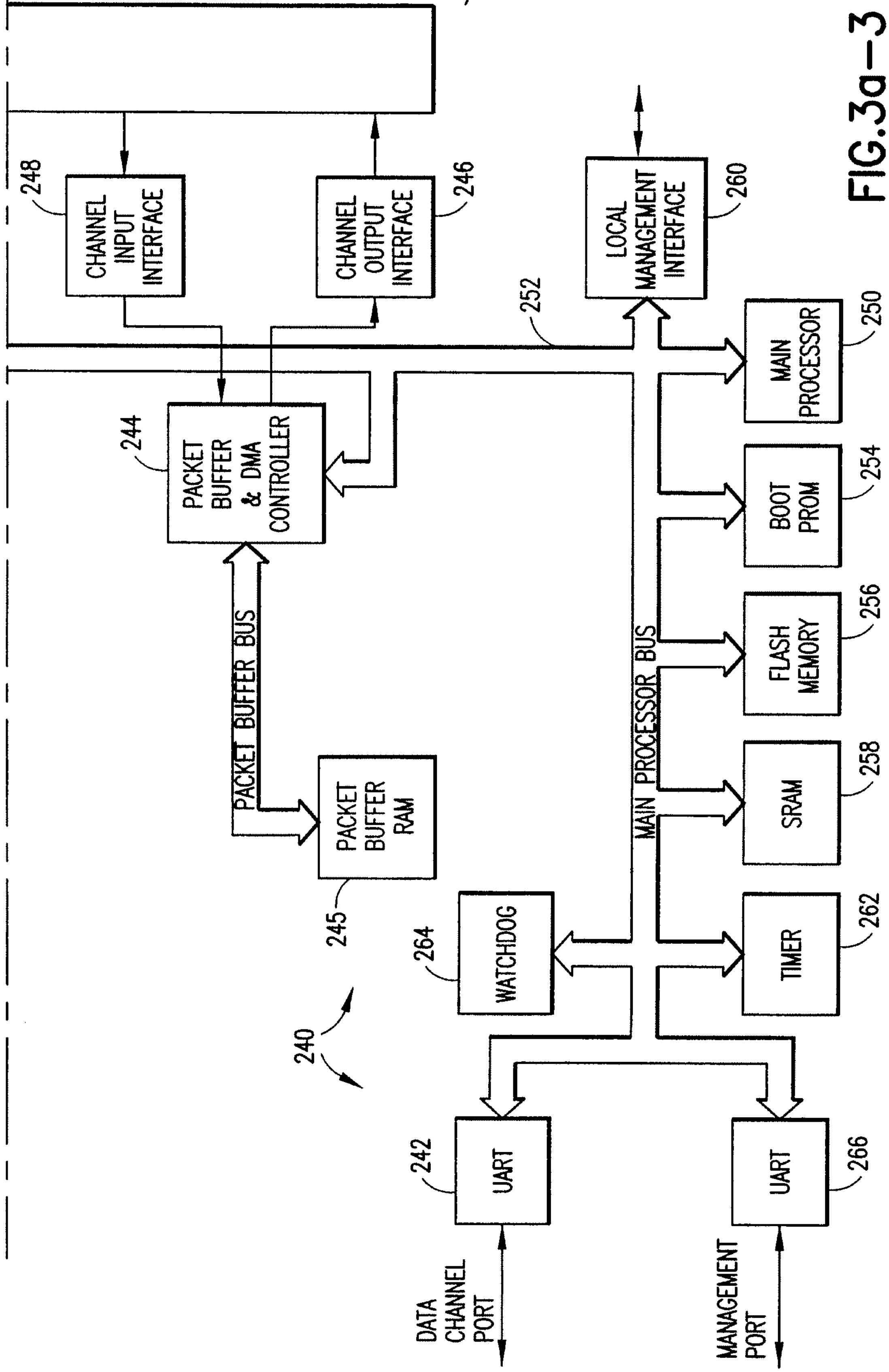
248
CHANNEL INPUT INTERFACE
246
CHANNEL OUTPUT INTERFACE
260
LOCAL MANAGEMENT INTERFACE
252
244
PACKET BUFFER & DMA CONTROLLER
PACKET BUFFER BUS
PACKET BUFFER RAM
245
250
MAIN PROCESSOR
254
BOOT PROM
256
FLASH MEMORY
258
SRAM
262
TIMER
MAIN PROCESSOR BUS
264
WATCHDOG
240
242
UART
266
UART
DATA CHANNEL PORT
MANAGEMENT PORT

FIG.3a–3

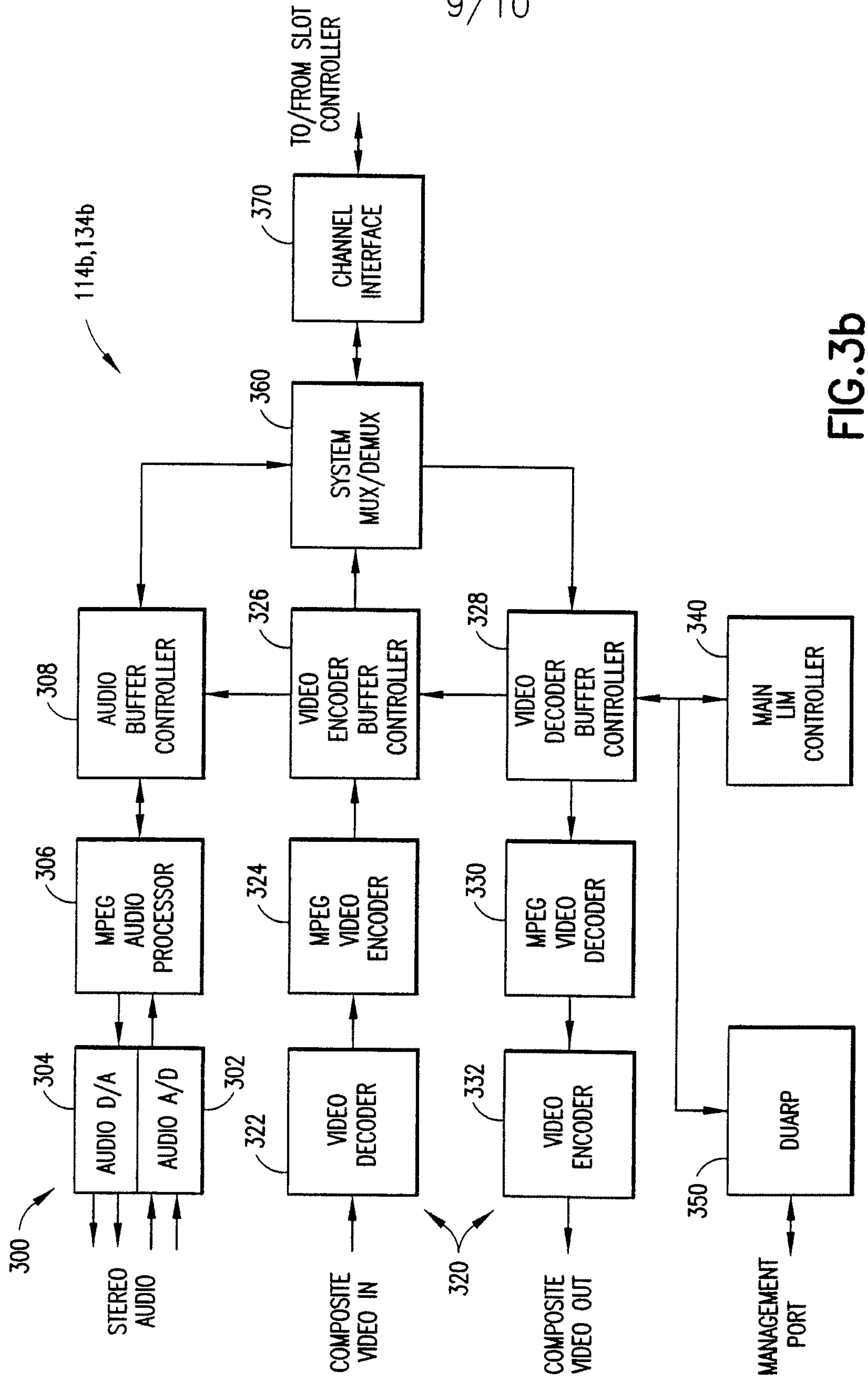
TO/FROM SLOT CONTROLLER
114b,134b
370
CHANNEL INTERFACE
360
SYSTEM MUX/DEMUX
308
AUDIO BUFFER CONTROLLER
326
VIDEO ENCODER BUFFER CONTROLLER
328
VIDEO DECODER BUFFER CONTROLLER
340
MAIN LIM CONTROLLER
306
MPEG AUDIO PROCESSOR
324
MPEG VIDEO ENCODER
330
MPEG VIDEO DECODER
304
AUDIO D/A
AUDIO A/D
302
322
VIDEO DECODER
332
VIDEO ENCODER
DUARP
350
300
STEREO AUDIO
COMPOSITE VIDEO IN
320
COMPOSITE VIDEO OUT
MANAGEMENT PORT

FIG.3b

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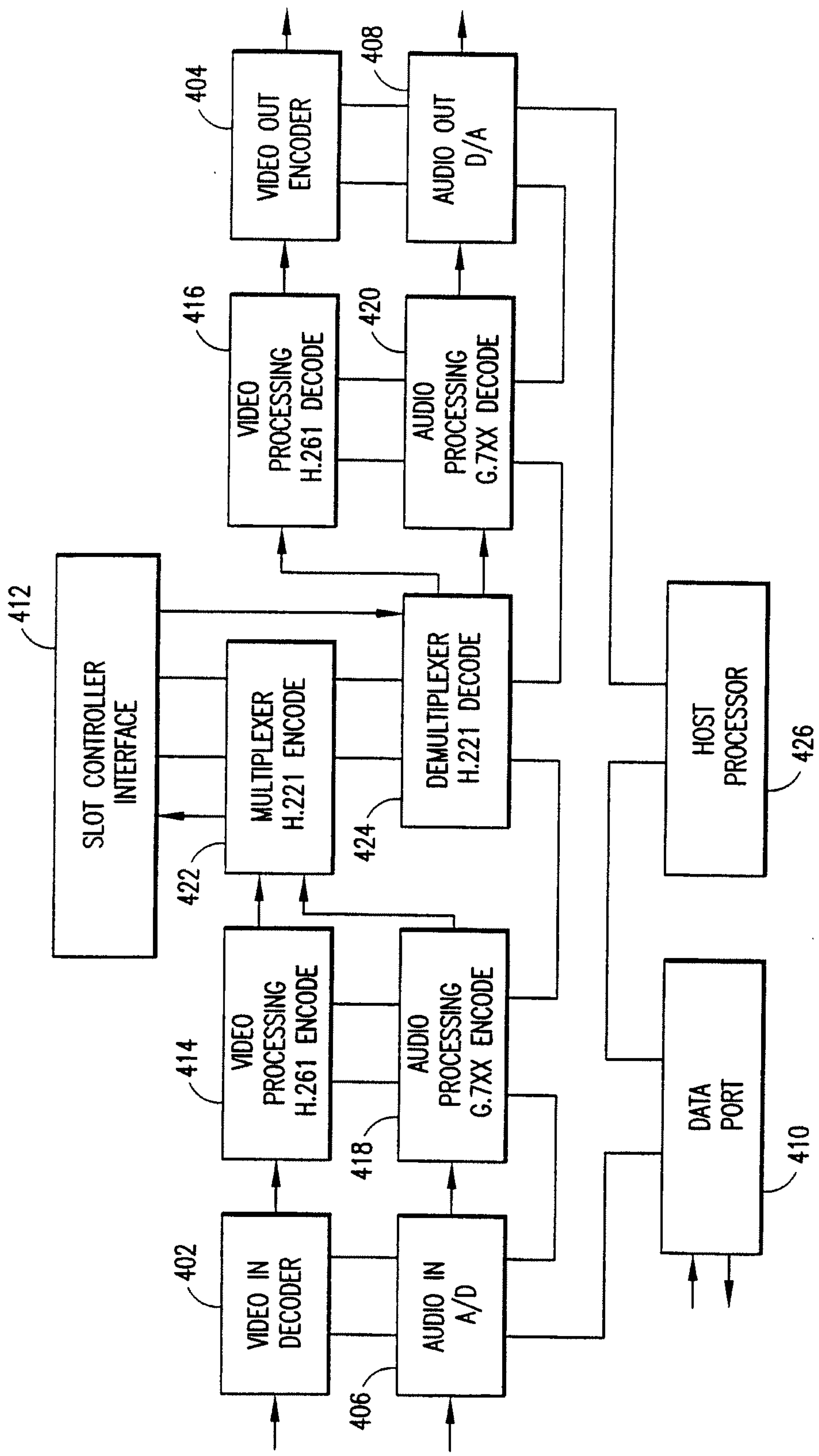
VIDEO IN DECODER
402
AUDIO IN A/D
406
VIDEO PROCESSING H.261 ENCODE
414
AUDIO PROCESSING G.7XX ENCODE
418
MULTIPLEXER H.221 ENCODE
422
SLOT CONTROLLER INTERFACE
412
DEMULTIPLEXER H.221 DECODE
424
VIDEO PROCESSING H.261 DECODE
416
AUDIO PROCESSING G.7XX DECODE
420
VIDEO OUT ENCODER
404
AUDIO OUT D/A
408
HOST PROCESSOR
426
DATA PORT
410

FIG.3c

INTERNATIONAL SEARCH REPORT

International application No.
PCT/US98/20706

A. CLASSIFICATION OF SUBJECT MATTER

IPC(6) :H04N 7/10, 7/14
US CL :348/12, 13, 441
According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

U.S. : 348/12, 13, 15, 17, 384, 385, 441; 395/200.34, 200.35, 200.36, 200.37

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X --- Y	WO 93/16557 A1 (KOZ et al) 19 August 1993, p.6, ln.17-31, p.7, ln.28, to p.9, ln.7, p.16, ln.16-37, p.17, ln.14-30, p.18, ln.30, to p.19, ln.6, p.24, ln.5-22, p.27, ln.5-25, p.28, ln.4-38, p.30, ln.16-38, and p.32, ln.32, to p.33, ln.2.	1-7, 10, 12-17 --------------- 8, 9, 11, 18-21
Y	US 5,555,017 A (LANDANTE et al) 10 September 1996, col.4, ln.19-23, and col.5, ln.23-25.	8, 9, 18-20
Y	US 5,488,433 A (WASHINO et al) 30 January 1996, col.4, ln.64-67.	11, 21

☐ Further documents are listed in the continuation of Box C. ☐ See patent family annex.

* Special categories of cited documents:
"A" document defining the general state of the art which is not considered to be of particular relevance
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"O" document referring to an oral disclosure, use, exhibition or other means
"P" document published prior to the international filing date but later than the priority date claimed
"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention
"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone
"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art
"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
10 DECEMBER 1998	27 JAN 1999

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